The Future of SIP

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Overview

- What is SIP good at?
- SIP and Corba
- on-going IETF SIP efforts
- SIP for presence and events
Principal IETF VoIP protocols

**RTP/RTCP**: data transport and QoS feedback

**SIP**: call setup

**SDP**: session/media description

**enum**: (DNS) E.164 → URLs

**TRIP**: finding “cheap” PSTN gateways, BGP-like

**RTSP**: voice mail, announcements
What is SIP good at?

- session setup = “out of band”
- resource location via location-independent identifier ("user@domain", tel)
- particularly if location varies rapidly or filtering is needed (i.e., is inappropriate for DNS and LDAP)
- real-time: faster than email
- reach multiple end point simultaneously or in sequence = forking
- possibly hide end-point location
- delayed final answer ("ringing") $\leftrightarrow$ RTSP
What is SIP not meant for?

- bulk transport: media streams, files, pictures, …
- asynchronous messaging ("email")
- resource reservation
- high-efficiency general-purpose RPC
### SIP and Corba

<table>
<thead>
<tr>
<th></th>
<th>SIP</th>
<th>Corba</th>
</tr>
</thead>
<tbody>
<tr>
<td>data</td>
<td>optional fields</td>
<td>versioning hard</td>
</tr>
<tr>
<td></td>
<td>two-level hierarchy</td>
<td>general, C-like</td>
</tr>
<tr>
<td>hiding</td>
<td>dynamic</td>
<td>directory-based</td>
</tr>
<tr>
<td>multiple</td>
<td>forking proxy</td>
<td>no</td>
</tr>
<tr>
<td>transport</td>
<td>UDP, TCP, …</td>
<td>TCP</td>
</tr>
<tr>
<td>strength</td>
<td>inter-domain</td>
<td>inter-domain</td>
</tr>
<tr>
<td>generality</td>
<td>session set-up</td>
<td>RPC, events, …</td>
</tr>
</tbody>
</table>

SIP servers can benefit from Corba *locally* for user location and service creation.
Current SIP efforts

• SIP to Draft Standard
• QoS and security preconditions
• inter-domain AAA and billing
• session timer for liveness detection
• early media (PSTN announcements)
• SIP for presence / instant messaging
• SIP-H.323 interworking
• reliable provisional responses
• DHCP configuration for finding SIP servers
• SIP for firewalls and NATs
• caller preferences
• services (transfer, multiparty calls, home)
• ISUP carriage
The near future of SIP

- move “RFC2543bis” to draft standard: clarifications, bug fixes

- SIP bake-off interoperability events:

<table>
<thead>
<tr>
<th>host</th>
<th>when</th>
<th>companies</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 Columbia University</td>
<td>April 1999</td>
<td>16</td>
</tr>
<tr>
<td>2 pulver.com</td>
<td>August 1999</td>
<td>15</td>
</tr>
<tr>
<td>3 Ericsson</td>
<td>December 1999</td>
<td>26</td>
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<tr>
<td>4 3Com</td>
<td>April 2000</td>
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<td>5 pulver.com</td>
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<td></td>
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<tr>
<td>6 dynamicsoft</td>
<td>December 2000</td>
<td></td>
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</table>
## SIP extensions

About 50 SIP-related current Internet drafts...

<table>
<thead>
<tr>
<th>extension</th>
<th>example</th>
<th>negotiation</th>
</tr>
</thead>
<tbody>
<tr>
<td>method</td>
<td>SUBSCRIBE, NOTIFY</td>
<td>OPTIONS, Allow</td>
</tr>
<tr>
<td>header</td>
<td>Session-Expires</td>
<td>Require</td>
</tr>
<tr>
<td>body</td>
<td>ISUP</td>
<td>Accept, Accept-*</td>
</tr>
<tr>
<td>status</td>
<td>183</td>
<td>treat by class</td>
</tr>
</tbody>
</table>

How to maintain backward compatability?

- can add ignore-if-you-like headers
- clients require capabilities via Require
- servers decline via Unsupported
- servers indicate capabilities via Supported
Coupling of signaling and QoS

- traditional (H.323) approach: use signaling to set up QoS
- but: separation of signaling and data flow
- SIP approach: security and QoS preconditions
- SDP in INVITE: “must have QoS”, don’t ring
- use any reservation protocol, then send PR-MET
SIP for single-line phones

- PacketCable: VoIP for cable modems
- primary line service
- NCS and DCS (distributed call signaling)
- DCS is based on SIP

- extensions:
  - “certified” caller ID
  - distributed call state
  - billing
  - media authorization
Event notification

• **SUBSCRIBE** to events (state changes) and then get **NOTIFY**

• originally, for fax events in PINT

• generic: “message waiting”, people presence, alarms . . .

• details remain to be worked out:
  
  – event as signaling or media stream?
  
  – how to describe events – headers, XML, . . . ?
  
  – state storage independent of user agents?
  
  – retrieve current state or just transitions?
Call control

Several sub-problems:

- transfer calls
  - = ask somebody to call a third party
  - originally, Also in BYE
  - proposal: TRANSFER
  - alternatives: PLEASE, with full SIP message in body or header

- multi-party call meshes
  - complicated due to distributed state
  - race conditions
  - not always needed: MCU, multicast

- third-party control —→ no extensions needed
Invisible Internet telephony

- currently: stand-alone application or PSTN phone
- chat applications
- distributed games
- virtual reality environments
- web pages and applets
- links in email messages
Integrating VoIP with the web

Everything linked together, with SIP redirecting and registering:

- tel: URLs
- email: send SIP via email, redirect calls to email
- web: links to and actual content (HTML, XML, audio clips, ...)
- chat and presence
- RTSP
Mobile Internet telephony

- user and terminal mobility are related
- mobile applications: mostly UDP (DNS, multicast) or short TCP transactions (SMTP, POP, IMAP)
- should make applications restartable
- little mobile-IP deployment
- use SIP to support mobile multimedia applications
- mobile IP and SIP mobility are complementary
Conclusion

- major protocol pieces in place
- operational issues: 911, anonymity, billing, OSS for services, …
- other uses: controlling toasters, light switches, …
- not just replicating existing telephone architecture and service
- programmability key to web success
- should become an invisible service
- need to keep low-end devices in mind (IPsec!)